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APPLICATION OF

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FOR UTILITY PATENT COVERING

METHOD AND APPARATUS FOR FOCUSING SOUND

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Patent Application of

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for

TITLE

Method and Apparatus for Focusing Sound

FEDERALLY SPONSORED RESEARCH

Not applicable

SEQUENCE LISTING OR PROGRAM

Not applicable

BACKGROUND OF THE INVENTION

Field of Invention

This invention relates to the field of sound, more specifically to methods of selectively transmitting sound from one source, while rejecting sound from other sources.

Description of Prior Art

We live in a noisy society. It is often necessary to identify and characterize sound from a source of interest, despite a din of competing sounds. The competing sounds do not have to be particularly loud to make this task difficult, as long as

they are of the same order of magnitude as those under study. Such is the case in medical stethoscopy, where contaminating sounds may obscure subtle signs of disease, and delay diagnosis and treatment.

Under most circumstances, assistive technology is not available, and we simply do our best to hear. We move closer to the source of the sound of interest, and away from other sound sources, taking advantage of the attenuation of sound with distance. We cock an ear toward the source, and plug the other ear, to benefit from the directional selectivity of a single ear. We cup a hand behind the ear to provide some amplification. And we mentally "tune in" to the sound's unique frequency distribution, to distinguish it from background noise.

Prior art technology uses the same basic techniques.

Distance Selectivity

Velocity or pressure-gradient microphones respond to particle velocity, thus differentiating between sounds from near and distant sources. Noise-cancelling microphones typically incorporate two transducers, spaced a few inches apart and connected in antiphase. Sounds which originate closer to one transducer attenuate en route to the farther transducer, and are only partially cancelled. Those coming from a distance affect both transducers more equally, and are more completely cancelled. Cancellation increases with wavelength, resulting in a falling bass response.

The two-transducer noise-cancelling method is subject to a comb filter effect. Frequencies which originate a whole number of wavelengths farther from one transducer than the other reach them in phase, and are completely cancelled.

Frequencies which reach the antiphase transducers 180° out-of-phase, in contrast, augment one another. Alternating peaks of cancellation and augmentation produce a comb filter effect.

Newer noise-cancelling microphones have a diaphragm which is fully exposed to sounds of interest on one side, and exposed to the environment through an aperture in the case on the other side. The *net* sound pressure on the diaphragm is converted to an electrical impulse. The size of the aperture is adjusted to maximize the cancellation of contaminating environmental noise.

Directional Selectivity

Early cone-shaped hearing aids provided some directional selectivity, as well as amplification. Sound waves traveling toward the user along the cone's axis were efficiently funneled into the ear canal, while those striking the cone at other angles were not. Cones, baffles, and reflectors are used in other applications to direct sound waves toward or away from a sensing point, and sound-absorbing materials are used to reduce unwanted noise. Good directionality is achieved with a parabolic acoustic mirror, which reflects sound to a microphone facing the mirror at its focal point. Output falls for wavelengths longer than the reflector diameter, or shorter than the diameter of the microphone diaphragm.

Microphones exhibit a directional response in their pickup characteristics. At low and middle frequencies, omnidirectional microphones are equally sensitive to sound in all directions. However, at wavelengths shorter than the diameter of the diaphragm, diffraction of sound to the rear of the microphone diminishes, and the unit becomes increasingly directional. Ribbon microphones are sensitive to sound

from the front and rear, but insensitive to sound from the sides. Cardioid microphones are sensitive to sound from the front, and far less sensitive to sound from the sides and rear. Like omnidirectional microphones, their directivity is greater at higher frequencies.

Zoom microphones are even more axis-selective. They consist of two cardioid microphones placed one behind the other and connected in antiphase. A frequency-dependent phase shifter is configured so that sounds from the front are reinforced, while side-propagated sound is cancelled. Directivity decreases at low frequencies.

Interference tube microphones have a similar polar response. These consist of an open-ended tube, with a series of holes or slots along one side, positioned in front of a microphone. The paths to the microphone, through the tube's open end and side fenestrations, are of equal length only for sound waves which originate along the tube's axis. Such waves arrive in phase, and are reinforced. Sound waves travelling these different paths from other sites arrive at the microphone out-of-phase, and are cancelled. The band of cancellation, and of high directivity, encompasses all frequencies with wave lengths more than double the tube length.

Laser interferometers can detect minute movements of reflective surfaces from great distances. In the absence of other causes, vibrations of such surfaces reflect the sound energy impinging on them. A laser beam may thus travel through a noisy environment, and be used to eavesdrop on a conversation taking place behind a closed window.

Frequency Selectivity

As noted, many of the devices and methods discussed above are selective only within certain frequency bands. When the frequency distribution of the sound of interest differs from that of competing sounds, the frequency responses of these devices and methods may be exploited to differentiate the sounds. In addition, filters may be used to pass selected frequencies, while rejecting others.

Combinations

Many devices use these techniques in combination. Cones or baffles are commonly used to direct sound from a selected source to a transducer. In some devices, one transducer is configured to primarily sense background noise, and its signal, or a (sometimes digitally-correlated) portion thereof, is subtracted from that of a second transducer. The resulting difference signal may then be amplified and filtered.

<u>Problems Identified in the Prior Art</u>

Despite improvements in the prior art, several problems remain:

(a.) The degree to which the prior art discriminates between sounds originating near a site of interest, and sounds originating farther away, is unsatisfactory for certain applications (*e.g.* conducting an interview in a noisy crowd).

(b.) The prior art primarily addresses sounds transmitted through a uniform medium (*e.g.* air) directly from their sources. These methods are not well suited for *selectively* sensing sounds *through* an interface (*e.g.* a patient's

skin).

(c.) The two-transducer noise-cancelling method is subject to a comb filter effect.

(d.) The selectivity of many of the devices and methods in the prior art is diminished or lost at certain frequencies. The distance selectivity of the two-transducer noise-canceling microphone, for example, is lost at low frequencies.

BRIEF SUMMARY OF THE INVENTION

Objects and Advantages

Accordingly, several objects and advantages of the present invention are:

(a.) to selectively transmit sound originating near a site of interest, while rejecting sound originating farther away, with greater discrimination than is possible with the prior art;

(b.) to selectively transmit sound from one source, while rejecting sound from other sources, even when the sensing means is separated from the sound sources by an interface;

(c.) to substantially reduce the comb filter effect;

(d.) to accomplish the foregoing objects throughout the frequency range of human hearing.

Further objects and advantages will become apparent from a consideration of the ensuing description and drawings.

Summary

The present invention is a method and apparatus for transmitting sounds originating close to a site, while sharply attenuating sounds originating farther away. A reference transducer is positioned as close as possible to the site under study. Three or more satellite transducers are positioned on a circle centered on the reference transducer, such that they divide the circle into equal arcs. The average signal of the satellite transducers, or a portion thereof, is subtracted from the signal of the reference transducer. The resulting difference signal may be amplified and/or filtered prior to being transduced to sound by speakers.

Subtraction of the signals results in a focusing effect. Sounds which originate close to the reference transducer are preserved, while those which originate farther away are sharply attenuated. The focusing effect increases modestly with frequency, but is strong even at 16 Hz, the lower limit of human hearing. Sounds which originate at predictable locations in relation to the transducers are eliminated almost completely, a feature which enables the user to listen to the sounds of two juxtaposed sound sources, one at a time.

The method disclosed offers greater distance selectivity at all frequencies than is possible using the prior art. It is suitable for use when the sounds to be discriminated are separated from the sensing means by an interface.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is a graphical illustration of the theoretical basis of the sound-focusing method described herein.

FIG. **2** is a graph which compares two-site signal subtraction to single-site sampling, in one dimension.

FIG. **3** is a logarithmic contour plot of two-site signal subtraction values for sounds of equal intensity originating in the *xy* plane of the sampling sites.

FIG. **4** is a logarithmic contour plot of the *ratio* of two-site signal subtraction to single-sampling-site values for sounds originating in the *xy* plane of the sampling sites.

FIG. **5** is an illustration of sinωt values at the sampling sites of a four-satellite configuration.

FIG. **6** is a logarithmic contour plot of **satellite-subtraction method (SSM)** values for sounds of equal intensity originating in the *xy* plane of the sampling sites of the four-satellite configuration of FIG. **5**.

FIG. **7** is a logarithmic contour plot of SSM values for sounds of equal intensity originating in the *xy* plane of the sampling sites of a three-satellite configuration.

FIG. **8** is a logarithmic contour plot of the *ratio* of SSM to single-sampling-site values for sounds of equal intensity originating in the *xy* plane of the sampling sites of the four-satellite configuration of FIG. **5**.

FIG. **9** is a logarithmic contour plot of SSM values in the *xz* plane of line DF,

for the four-satellite configuration of FIG. **5**.

FIG. **10** is a diagram of the compound microphone of the preferred embodiment of the four-satellite configuration of FIG. **5**.

FIG. **11** is an electrical schematic diagram of the preferred analog embodiment of the four-satellite configuration of FIG. **5**.

FIG. **12** is a circuit board diagram of the preferred analog embodiment of the four-satellite configuration of FIG. **5**.

FIG. **13** is an electrical schematic diagram of a simple analog embodiment of the SSM.

DETAILED DESCRIPTION OF THE INVENTION

Theoretical Basis of the Invention

The **satellite-subtraction method (SSM)** described herein sharply discriminates between sounds which originate near its reference sampling site, and sounds which originate farther away. To accomplish this, the SSM exploits characteristics of the attenuation of sound with distance.

The intensity of sound is inversely proportional to the square of the distance from its source:

$$I \propto 1/d^2 \quad \text{or} \quad I = K/d^2$$

where I is intensity, d is distance, and K is the proportionality constant. Thus the *relative* intensities 1, 2, and 3 units from the source are $1/1^2$, $1/2^2$, and $1/3^2$, respectively.

First consider the issue in **one dimension** (FIG. 1). The distance from a sound source S_1 at x=-2 to a sampling site at x is $|-2-x| = x+2$ units. The solid curve plots the relative intensity $I_1 = 1/(x+2)^2$ of sound from S_1 at sampling sites from x=-2 to x=1.5. Similarly, for a second sound source S_2 at x=-1, the dashed curve plots the intensity function $I_2 = 1/(x+1)^2$. At a single sampling site B (x=0), the relative intensity of the nearer sound (dashed curve) is $1/(0+1)^2 = 1$, while that of the more distant sound (solid curve) is $1/(0+2)^2 = 0.25$. At B, the intensity of the nearer sound is therefore 1/0.25 = 4 times that of the more distant sound. Thus the attenuation of sound due to *distance alone* substantially differentiates near and distant sounds.

Two-Site Signal Subtraction

Now consider two sampling sites (FIG. 1), a **reference site** A (x=-0.5) and a **satellite site** C (x=+0.5). The intensity of the nearer sound (I_2, dashed curve) at A is $1/(-0.5+1)^2 = 4$; at C it is $1/(0.5+1)^2 = 0.44$. The difference ΔI_{2AC} between the nearer sound's intensities at A and C is 4 - 0.44 = 3.56. Subtraction thus preserves 3.56/4 = 89% of the nearer sound's intensity at A. Similarly, the difference ΔI_{1AC} between the more distant sound's intensities (I_1, solid curve) at the two sampling sites is 0.44 - 0.16 = 0.28. Subtraction preserves only 0.28/0.44 = 64% of the distant sound's intensity at A.

It is clear that the **two-site signal subtraction (TSSS)** illustrated in this example treats near and distant sounds differently. The ratio of the differences, 3.56/0.28 = 12.5, is substantially more than the ratio (4) of the intensities sampled at the single site B. It may be said that TSSS discriminates between the near and

distant sounds selected for this example 12.5/4 = 3.125 times as well as single site sampling.

Thus far, we have considered only sound sources located at x=-2 and x=-1. Let us now derive a general formula for the discriminatory improvement afforded by TSSS. In general, the distance from a sound source S (at x_S) to a sampling site B (at $x=0$) is x_S units, and the sound's relative intensity at B is $I_B = 1/x_S^2$. Similarly, the sound's relative intensity at A ($x=-0.5$) is $I_A = 1/(x_S - -0.5)^2 = 1/(x_S+0.5)^2$, and at C ($x=+0.5$) it is $I_C = 1/(x_S-0.5)^2$. The difference ΔI_{AC} between the sound's intensity at A and C is $\Delta I_{AC} = I_A - I_C = 1/(x_S+0.5)^2 - 1/(x_S-0.5)^2$. The *ratio* of this difference to the intensity at B is $\Delta I_{AC}/I_B = [I_A-I_C]/I_B = [1/(x_S+0.5)^2 - 1/(x_S-0.5)^2]/(1/x_S^2)$. This is graphed in FIG. 2.

Note that sounds which originate equidistant from A and C (*i.e.* at the **midpoint** of $\overline{AC}$, at B ($x = 0$) in this case) are equally intense at both sites ($I_A=I_C$), and are completely eliminated by subtraction ($[I_A-I_C]/I_B = 0$). Other than this dip in the ratio at $x=0$, it is clear that, compared to the attenuation of sound due to distance alone, TSSS accentuates sounds which originate close to either site (ratio > 1 near A and C), while attenuating sounds which originate farther away (ratio < 1). It is clear that, in one dimension, two-site signal subtraction improves substantially upon the attenuation of sound with distance, providing a focusing effect.

Now consider the issue in **two dimensions**. The *distance* d_{SA} from a sound source S, located at (x_S,y_S), to a **reference** sampling site A, located at (x_A,y_A), is $d_{SA} = [(x_S-x_A)^2+(y_S-y_A)^2]^{1/2}$. Since the *intensity* of a sound at a sampling site varies inversely with the square of its distance from the sound's source, the sound's relative intensity

at A is $I_A = 1/(d_{SA})^2 = 1/([(x_S-x_A)^2+(y_S-y_A)^2]^{1/2})^2 = 1/[(x_S-x_A)^2+(y_S-y_A)^2]$. The difference ΔI_{AC} between the sound's intensity at A and at a **satellite** sampling site C (x_C,y_C) is $\Delta I_{AC} = I_A - I_C = 1/[(x_S-x_A)^2+(y_S-y_A)^2] - 1/[(x_S-x_C)^2+(y_S-y_C)^2]$. FIG. 3 maps ΔI_{AC} for sound sources of equal intensity positioned with respect to sampling sites A (-0.5,0) and C (+0.5,0) in a uniform medium. Note the sharp drop-off in ΔI_{AC} with distance from A or C.

The pattern is the same for any plane which includes line AC. Thus the graph in **three dimensions** is simply a rotation of FIG. 3 about line AC. Sounds which originate in the plane equidistant from A and C (the "**midplane**", $x = 0$ in FIG. 3) are equally intense *and in phase* at both sampling sites, and cancel one another completely during subtraction ($\Delta I_{AC} = 0$).

FIG. 4 (a logarithmic contour plot of $\Delta I_{AC}/I_B$) compares the extent to which TSSS discriminates between near and distant sounds, to that of a single sampling site B (0,0), positioned midway between TSSS sites A (-0.5, 0) and C (+0.5, 0). Note that TSSS is *both* more efficient at preserving nearby sounds (ratios > 1) *and* attenuating distant sounds (ratios < 1), compared to the single site.

For any **reference** sampling site A and **satellite** sampling site C, both the absolute *and relative* differences between a sound's intensity at the sites decrease as its source moves away (FIGS. 1 and 3). TSSS exploits this to preserve sounds originating close to either site, while attenuating sounds originating farther away.

Two-site signal subtraction has its shortcomings, however, and they relate to the **wave nature of sound**. If v is the speed of sound, a sound which originates a distance Δd feet closer to one sampling site than the other will reach it $\Delta d/v = \Delta t$

seconds sooner. For sources in the midplane (equidistant from sites A and C), $\Delta d = 0$ and $\Delta t = 0$. Sounds which originate in the midplane therefore always reach both sampling sites simultaneously, and in phase.

In all other cases, $\Delta d \neq 0$. If Δd is a multiple n of a sound's wavelength λ, $\Delta d = n\lambda$, and $\Delta d/\lambda = n$. When a sound source is a whole wavelength, or a whole multiple thereof, nearer one sampling site than the other, n is a whole number. Sounds of these wavelengths reach the sampling sites in phase, and are subtracted in phase, resulting in cancellation. Sounds for which $n+0.5$ is a whole number reach the sites 180° out-of-phase, and are subtracted in antiphase. Subtraction of antiphase signals is equivalent to *adding* in-phase signals. Alternating peaks of cancellation and augmentation produce a frequency-related **comb filter effect**.

A *spatial* comb filter effect also plagues the two-site subtraction method. As a source emitting sound of a given wavelength is moved so that Δd changes, $n = \Delta d/\lambda$ changes too. Cancellation is maximal when n is a whole number, and augmentation when $n+0.5$ is a whole number. This produces a pattern of peaks and valleys throughout the sound field for each frequency.

TSSS can be used to dissect a composite of sounds into its components. The first heart sound, for example, is comprised of the closure sounds of the tricuspid and mitral valves. If sampling site A is placed directly over the tricuspid valve, and sampling site C is positioned so that the mitral valve is in the midplane equidistant from A and C (*i.e.* straddled by them), the sound of the mitral valve is subtracted from the composite. The procedure is reversed to subtract the sound of the tricuspid valve.

Satellite-Subtraction Method

It is possible to improve upon two-site signal subtraction, virtually eliminating both its comb filter effect and falling bass response, while substantially increasing distance selectivity, by adding **additional sampling sites**. A **reference** sampling site (positioned as close as possible to a sound source of interest) is designated, and n ($\geq$3) **satellite** sampling sites are positioned on a circle centered on the reference site, such that they divide the circle into equal arcs. Put another way, n ($\geq$3) satellite sampling sites are positioned at the vertices of an n-sided regular polygon centered on the reference site. The average of the sound intensities sampled at the satellite sites is subtracted from the sound intensity at the reference site, $I_{ref} - \Sigma I_{sats}/n = \Delta I_{n+1}$.

A more detailed analysis of the **wave nature of sound** is appropriate at this point. Like any periodic waveform, sound can be represented as the sum of sine waves having different amplitudes, frequencies (f), and phase relations. The amplitude L of any one such sine wave at a given sampling site P at a moment in time t is $L_{P(t)} = I_P \sin \omega t$, where ω, the angular velocity of the sine wave in radians per second, is equal to $2\pi f$. The wave travels from its source S (x_S, y_S) to P (x_P, y_P) in $t = d_{SP}/v = [(x_S-x_P)^2 + (y_S-y_P)^2]^{1/2}/v$ seconds. Then $L_{P(t)} = I_P \sin \omega t = (1/[(x_S-x_P)^2+(y_S-y_P)^2])\sin([2\pi f][(x_S-x_P)^2+(y_S-y_P)^2]^{1/2}/v)$.

Adapting the equation $I_{ref} - \Sigma I_{sats}/n = \Delta I_{n+1}$ to include this sinusoidal variation of sound **wave amplitude**, it becomes $L_{ref(t)} - \Sigma L_{sats(t)}/n = \Delta L_{n+1(t)}$. Since all sites are sampled *simultaneously*, the equation for the time it takes sound to travel from its source to the *reference* site is used in each term of this equation.

Consider, for example (FIG. 5), a reference sampling site R (0,0) **20**, and four satellite sampling sites D (-1,0) **22D**, E (0,1) **22E**, F (1,0) **22F**, and G (0,-1) **22G** positioned on a circle **24** centered on R, such that they divide the circle into equal arcs. Sound spreads outward from a source S **26** in all directions (arrows). The value of sin ωt at a moment in time is illustrated for each sampling site. (The *declining peak-to-peak value* I of each sine wave, which is inversely proportional to the square of the linear distance *d* from S, is *not* illustrated.) The amplitude of the sound wave at a given sampling point at a moment in time is given by $L_{P(t)} = I_P \sin \omega t$. For this example, the pertinent values are:

Sampling Site	*d* (from S)	$I_P = 1/d^2$	$\sin\omega t$	$L_{P(t)} = I_P \sin\omega t$
R	1.5	0.444	-1.0	-0.444
D	0.5	4.0	1.0	4.0
E	1.80	0.308	-0.581	-0.179
F	2.5	0.16	1.0	0.16
G	1.80	0.308	-0.581	-0.179

and $L_{R(t)} - \Sigma L_{sats(t)}/4 = \Delta L_{5(t)} = -1.395$ for the moment in time illustrated.

This example was chosen as a worst-case scenario: S emits sound of wavelength $\lambda = 2d_{DR}$ from a location one-fourth wavelength outside $\overline{DF}$ on line DF. If only reference site R and satellite sites D and F were present, an augmentation peak of $\Delta L_{3(t)} = -0.444 - (4.0 + 0.16)/2 = -2.524$ would be produced. The addition of sites E and G adds negative terms to $\Sigma L_{sats(t)}$, reducing the peak to -1.395. For comparison, the augmentation peak produced by a prior art two-transducer (D and F) noise-cancelling microphone (embodying TSSS) in a similar worst-case scenario (180°

phase mismatch) is 4.16. This illustrates how the addition of satellites substantially reduces the comb filter effect seen with TSSS.

FIG. 6 is a logarithmic contour plot of the **absolute wave amplitude** $|\Delta L_{5(t)}|$ for 200 Hz sound sources with a phase angle of $\pi/2$, when $v = 1{,}545$ m/s (the speed of sound in water at body temperature) and one distance unit equals 2.5 cm. The dashed curves indicate where $|\Delta L_{5(t)}| = 1.0$. (Similar conditions and conventions were used for FIGS. 7-9.) Note the conservation of sounds originating near sampling sites D, E, F, G, and R (larger $|\Delta L_{5(t)}|$), and the sharp attenuation of sounds originating farther away (smaller $|\Delta L_{5(t)}|$).

The contours exist in relation to the sampling sites; doubling the distance unit and wavelength modeled results in an identical plot. The pattern shrinks slightly as frequency increases. (For comparison, FIG. 7 is a similar plot of $|\Delta L_{4(t)}|$.) FIG. 8 plots the *ratio* of $|\Delta L_{5(t)}|$ to the absolute wave amplitude sampled at the single site R, $|L_{R(t)}|$. Note that, compared to single site sampling, the **satellite-subtraction method (SSM)** is far more efficient at attenuating distant sounds (ratios < 1). Sounds which originate very close to the sampling sites, in contrast, are preserved (ratios ≥ 1).

The focusing effect illustrated in two dimensions in FIG. 6 actually occurs in three dimensions. FIG. 9 is a logarithmic contour plot of $|\Delta L_{5(t)}|$ in the *xz* plane of line DF, using the same sampling sites as FIGS. 5, 6, and 8. (The plot is similar in the *yz* plane of line EG.) Note that when $n=4$ in $L_{ref(t)} - \Sigma L_{sats(t)}/n = \Delta L_{n+1(t)}$ sounds which originate very close to the sampling sites (especially the reference site) are preserved, while those originating farther away are sharply attenuated.

In certains situations, a degree of distance selectivity *less than* that illustrated

may be desirable. Dividing $\Sigma L_{sats(t)}$ by a number other than n changes the focusing effect; increasing the divisor to infinity makes $|\Delta L_{n+1(t)}|$ equal to $|L_{ref(t)}|$, the absolute sound wave amplitude sampled at the single site R. Embodiments which allow the user to continuously vary the divisor provide a continuously variable focusing effect.

Note also in FIGS. 6 and 9 that the zone of cancellation **28** (where $|\Delta L_{5(t)}| \approx 0$), no longer a simple "midplane", has taken on a complex shape. Nevertheless, its proximity to the focal zone (higher values of $|\Delta L_{5(t)}|$) makes it possible to listen to the sounds of two juxtaposed sound sources, one at a time. To do so, position the sensing sites so that the sound source under study is in the focal zone, and the source to be eliminated is in the zone of cancellation.

Thus the satellite-subtraction method preserves sounds which originate close to its sampling sites, while sharply attenuating sounds which originate farther away. It provides greater distance selectivity, throughout the frequency range of human hearing, than was possible with the prior art. It allows the user to listen to the sounds of two juxtaposed sound sources, one at a time. Compared to two-site signal subtraction, it provides a sharper, more targeted, and continuously adjustable focus.

Embodiments

Embodiments of the satellite-subtraction method have acoustic-electrical transducers (*e.g.* microphones) at the sensing sites, circuitry which accomplishes the required signal processing, and accommodation (*e.g.* speakers or a connector suitable for speakers) for converting the resulting signal to a form suitable for interpretation. Allowing for the fact that transducers, unlike the sampling sites of the foregoing

discussion, have area, the performance of such embodiments, under test conditions in which sound travels in a uniform medium, matches the modeled values. Although only three- and four-satellite configurations were modeled (FIGS. 6 and 7), additional satellite sites may be added, consistent with the method described.

The satellite-subtraction method is suitable for use alone, or in combination with prior art. Cones, baffles, and sound-insulating materials may be used to direct sound from a selected source to or away from a given transducer. Transducers should be selected which are suitable for the sensing environment (*e.g.* microphones or hydrophones) and the frequencies under study; it is preferred that the satellite transducers be functionally equivalent. The modeling illustrated in figures referenced herein assumes transducers with an omnidirectional pickup pattern, but in some circumstances transducers with other pickup patterns (*e.g.* cardioid) may be more advantageous.

Averaging the signals of the satellite transducers may be accomplished by analog and/or digital addition and division, or by simply connecting them in parallel between a direct current-biased signal wire **30** and system ground (FIG. 11). Filters **32** may be used to shape the SSM's overall frequency response.

Preferred Four-Satellite Embodiment

The preferred four-satellite embodiment of the SSM is illustrated in FIGS. 10-12. FIG. 10 depicts a *compound* microphone consisting of five microphones in a housing **34** on a circuit board **36**, and a section through microphones E, R, and G. Satellite microphones (also referred to more generally as satellite transducers) D **38D**, E **38E**, F **38F**, and G **38G** are positioned on a circle **24** centered on reference

microphone (also referred to more generally as reference transducer) R **40**, and divide the circle into equal arcs. A funnel-shaped feature **42** of the housing surrounds, and directs sound to, reference microphone R, while a common trough **44** directs sound to the satellite microphones. Circular **46** and annular **48** diaphragms (not shown in the *en face* view) with appropriate resonant frequencies may optionally be included to facilitate sound transfer (*e.g.* at an interface) and to protect the transducers.

A 9-volt battery **50** serves as a source of electrical power (FIG. 11). Identical satellite microphones **38D-G** are connected in parallel between a direct current-biased signal wire **30** and ground, an arrangement which automatically averages their signal voltages (*i.e.* calculates $\Sigma L_{sats(t)}/4$). The direct current bias voltage is set to the standard operating voltage of the satellite microphones by adjusting a satellite trimmer potentiometer **52**. The reference microphone **40** is similarly connected between its standard operating voltage and ground. Since all of the microphones are connected directly to the circuit board **36** (FIGS. 10 & 12), electromagnetic interference is minimized, and neither balanced microphones nor screened cables are required.

Large-value input capacitors **54** are used to strip the reference and satellite signals from their direct current bias voltages, and voltage dividers **56** are used to rebias the signals midway between V+ and V-. Unfortunately, capacitors attenuate signal components of very low frequency. In some applications, such as medical stethoscopy, these frequencies are often the focus of study. (The dominant frequency of the first heart sound, for example, is just 22.6 ± 9.6 Hz.) It is

important to note, therefore, that the capacitors (and rebiasing circuitry) may be omitted if the signal wires are biased to half the system voltage.

An operational amplifier **58** applies a fixed gain of ten to the reference signal, and a focusing control **60** allows the user to apply a gain of zero to ten to the satellite signal. An instrumentation amplifier (INA) **62** is used to subtract the amplified satellite signal from the amplified reference signal, and a volume control **64** determines how much gain is applied to the resulting difference signal. An INA trimmer potentiometer **66** is adjusted to set the reference voltage of the intrumentation amplifier, so that the direct current bias voltage of its output signal is midway between V+ and V-.

A high-order low-pass filter **32**, the corner frequency of which the user can select via a filter control **68** (which determines the output frequency of a relaxation oscillator **70**), removes high-frequency contaminants from the signals of interest. The frequencies of normal heart sounds, pathologic heart sounds, physiologic fundamentals, and physiologic harmonics fall below 200, 800, 3000, and 6000 Hz, respectively. When corresponding corner frequencies are selected, the MAX291 8th-order low-pass Butterworth filter **32** removes 99, 96, 85, and 70 percent, respectively, of amplified Johnson noise, while preserving the sounds of interest. (A Bessel filter is less efficient at removing Johnson noise, but it maintains the phase relationships of signal components, and may be the better choice when it is essential that those relationships be preserved.)

In applications in which a fixed corner frequency is acceptable, the relaxation oscillator **70**, and its supporting circuitry, may be replaced with a single capacitor.

(See the MAX291 data sheet.) A fixed corner frequency of 1544 Hz, the high note of a soprano, to 3088 Hz, the second harmonic of that note, would be appropriate in a SSM embodiment configured for voice. A power amplifier **72** passes the filter's output to an external device (*e.g.* headphones) via an output connector **74**. Noise-canceling headphones may be used in high-noise environments.

Operation of the Preferred Four-Satellite Embodiment

In order to benefit from the advantages of the SSM, users of the preferred four-satellite embodiment must adjust its three thumbwheel potentiometer controls **60 64 68** appropriately. The headphone ear cups are placed over the ears, and the compound microphone (FIG. 10) is positioned close to, and facing, the sound source of interest. It is recommended that the corner frequency (filter control **68**, FIGS. 11 and 12) initially be set high, and the focusing control **60** low, so that the sounds presented to the user are most inclusive. The volume control **64** should also initially be set low. The momentary-on power switch **76** is then closed, and the volume is increased to a comfortable level. The user selects particular sounds for further study, and repositions the compound microphone so that its central reference microphone **40** is as close as possible to the source of the sound of interest. The focusing power, corner frequency, and volume may then be adjusted as desired. In general, the volume should be decreased *prior to* any increase in the corner frequency or decrease in the focusing power.

To listen to the sounds of two juxtaposed sound sources, one at a time, position the compound microphone so that the sound source under study is in the reference microphone's focal zone, and the source to be eliminated is in its zone of

cancellation **28** (FIG. 9).

Simple Four-Satellite Embodiment

A simple analog four-satellite embodiment of the SSM, with a fixed focus and no filter, may be constructed as diagrammed in FIG. 13. (Primes of the reference numerals used in the preferred embodiment are used for corresponding components in the simple embodiment.) This configuration may be satisfactory for use in a broadcast journalist's microphone, the signal of which will be processed further at the broadcast facility. Note that FIG. 13 further illustrates an embodiment with no capacitors in the signal path, a feature which eliminates the attenuation of low-frequency signal components due to series capacitance.

Applications

The advantages of the SSM suggest a variety of possible applications. In noisy environments, where excellent distance selectivity is desirable, a simple embodiment with a fixed focus and a fixed filter corner frequency (or no filter at all), would improve speech reception and noise rejection. Journalists conducting interviews (and ordinary people using cellular telephones) in noisy crowds, helicopter crews, and combat radiotelephone operators are among possible beneficiaries of this application. (Noise-cancelling headphones may be used in combination with the SSM in particularly noisy environments.)

When sounds of low frequency and low intensity, separated from the sensing means by an interface, are to be discriminated (as in medical stethoscopy), the preferred embodiment is appropriate. The radius of the circle **24** on which the satellite microphones are positioned must be carefully selected, so that the

stethoscope has enough depth of focus (FIG. 9), but is not unwieldy. A radius of approximately 3 to 5 cm meets these requirements. Round and annular diaphragms with appropriate resonant frequencies improve sound transfer at the skin interface and protect the microphones. Ideal microphone characteristics are high sensitivity and signal-to-noise ratio, low current consumption, and a flat frequency response down to 16 kHz. Microphones which may be operated at half the system voltage make it possible to omit the input capacitors, limiting the undesirable attenuation of clinically-important signal components of very low frequency. If input capacitors *must* be used, they should be of high value (*e.g.* 470 μF). The circuit diagrammed in FIG. 11, with thumbwheel potentiometers (FIG. 12) for the focusing **60**, volume **64**, and filter **68** controls, may be used. The output connector **74** should be a stereo audio jack, and it is preferred that high-quality headphones be used with the stethoscope.

Conclusion

All of the electronic components illustrated in the drawings are commercially available. FIGS. 11 and 13 depict the use of AD620 instrumentation amplifiers (manufactured by Analog Devices Incorporated of Norwood MA), a MAX291 low-pass filter (manufactured by Maxim Integrated Products of Sunnyvale, CA), LM386 power amplifiers (manufactured by National Semiconductor Corporation of Arlington TX), and TLC2272 and TLV2772 operational amplifiers (manufactured by Texas Instruments of Dallas, TX). One skilled in the art will recognize that other components may be substituted.

While the above description contains many specificities, these should not be construed as limitations on the scope of the invention, but rather as exemplifications of a few possible variations thereof. Many other variations are possible. Accordingly, the scope of the invention should be determined not by the examples illustrated, but by the appended claims and their legal equivalents.